



Metaswitch VP2510 Integrated Softswitch: Connecting Cisco Unified Communications Manager 8.6 via the Cisco Unified Border Element 8.7 using SIP



7/12/2012

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Introduction

Service Providers today, such as Metaswitch, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. Metaswitch is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 8.6 with a Cisco Unified Border Element (Cisco UBE) 8.7 for connectivity to Metaswitch VP2510 Integrated Softswitch. The deployment model covered in this application note is CPE to PSTN. This document does not address 911 emergency outbound calls. For 911 feature service details contact Metaswitch, directly.
- Testing was performed in accordance to Cisco's SIP Trunk Test Plan and all features were verified. Key features verified are:
 - CPE outbound to SP Offnet gateway (PSTN) (G.729 is offered first)
 - SP offnet gateway (PSTN) inbound to CPE (G.729 offered first)
 - CPE to CPE (place call out to the SP network and back) (G.729 is offered first)
 - CPE Calling number privacy
 - CPE Telephone Number Support – digit translations
 - CPE Calling Name Delivery
 - CPE offnet Call Conference
 - CPE Intra-Site Call Conference
 - CPE Intra-Site Attended Call Transfer
 - CPE Intra-Site Unattended Call Transfer
 - CPE Intra-Site Blind Call Transfer
 - CPE Call Hold and Resume (call hold is always done on the IP PBX side)
 - CPE Voice Mail
 - SP Voice Mail
 - CPE Find Me (CFU)
 - CPE T.38 FAX SG3 (G.729 is offered first)
 - Simultaneous Calls
 - CPE Auto Attendant
 - CPE to PSTN Offnet gateway international call
 - CPE G.711 FAX SG3
 - CPE Find Me (Call Forward On Busy)
 - CPE Find Me (Call Forward Don't Answer)
 - Codec mid-call re-negotiation (to be tested without transcoder)

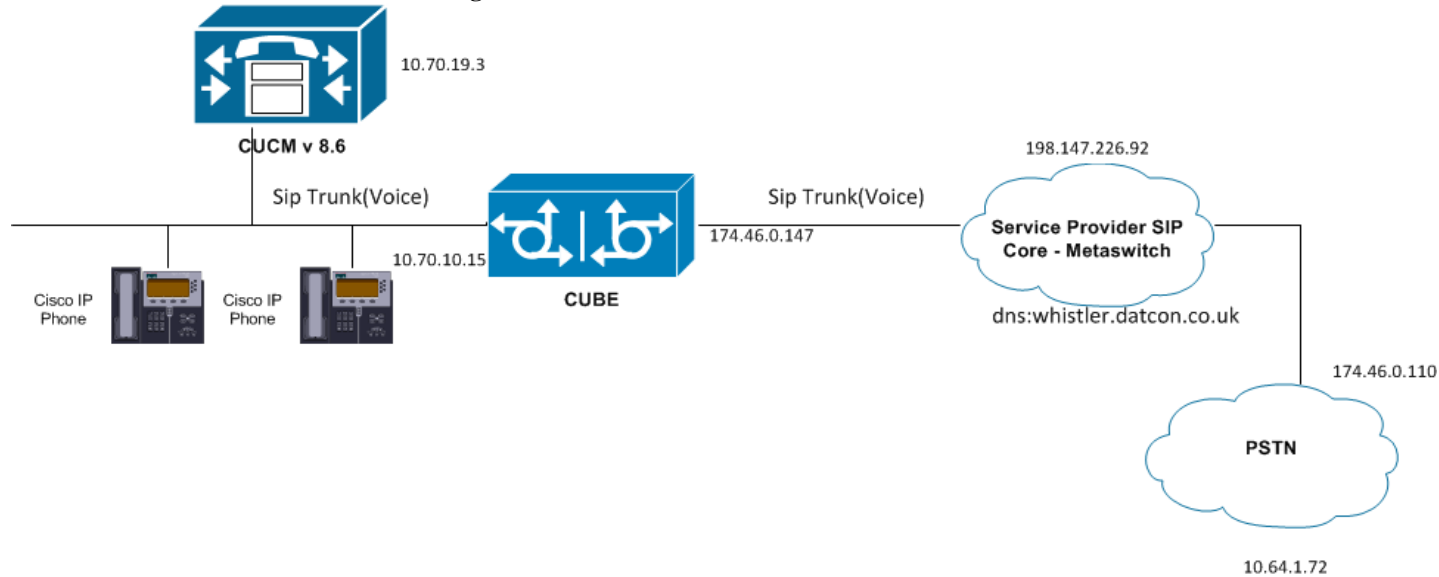


- PRACK with SDP (early-media cut-through with DTMF (RFC2833) navigation before 2000OK) - call 800-864-8331 - United Airlines
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Metaswitch SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to Metaswitch SIP network.
- This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucm/admin/8_0_2/ccmsys/a03ptcss.html

Network Topology

Figure 1 Basic Test Environment



System Components

Hardware Components

Component	Qty	Hardware	Notes
Call Manager	1	MCS7835	
CUBE	1	CISCO3945	
PSTN GW	1	C3845	To handle PSTN calls
Switch	1	C6509	Lab Switch
Cisco Phones	4	7962,9971,	Cisco Phones used as CPE.
Metaswitch VP2510 Integrated Softswitch	1	-	Third Party SIP Trunk Provider

Software Requirements

Component	Qty	Software Version	Firmware Version
Metaswitch VP2510 Integrated Softswitch	1	V7.3.00 SU56	
CUBE (10.70.10.15)	1	c3900e-universalk9-mz.SPA.151-4.M4.bin	
Communications Manager	1	8.6.1.20000-1	
Local PSTN GW	1	C3845-advipservicesk9-mz, Version 12.4(24)T	
Main Switch	1	Cisco WS-C6509 s3223-adventerprisek9_wan-mz, Version 12.2(33	
Cisco Phone 7960 (x2644)	1	P0S3-8-12-00	
Cisco Phone 9971 (x2700)	1	sip9971.9-1-1SR1	
Cisco Phone7962(x2604)	1	SCCP42.9-2-1S	



Features

Features Supported

- Call from/to PSTN to/from CPE – Basic and International calls , digit translations
- Hold/Resume
- DTMF
- Call transfers – attended, blind, unattended
- Call Forwarding (CFU,CFB,CFNA)
- FAX : T.38 and G711ulaw
- Support for early media

Features Not Supported

- Connected party update



Caveats

- As connected party update is currently not supported by Metaswitch, CLID updates are not observed on call transfer scenarios.
- Metaswitch doesn't allow T.38 to T.38 or T.38 to G711 calls to traverse their UMG. Therefore the FAX calls were tested with media flowing directly from the CUBE to the PSTN gateway.
- During the call transfer scenarios, no ring back is heard on the transferring party when a transfer is initiated i.e : When A and B are in call and B initiates a transfer to C, B does not hear a ring abck when C is ringin g. Even afetr B transfers the call in te case of an unattended transfer, A doesn't hear the ring back.



Configuration

Configuring Cisco Unified Border Element

Show version:

```
CUBE#      sho ver
Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.1(4)M4,
RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2012 by Cisco Systems, Inc.
Compiled Tue 20-Mar-12 22:07 by prod_rel_team
```

```
ROM: System Bootstrap, Version 15.1(1r)T4, RELEASE SOFTWARE (fc1)
```

```
CUBE uptime is 19 hours, 16 minutes
System returned to ROM by reload at 18:52:52 UTC Mon Jun 25 2012
System restarted at 18:56:47 UTC Mon Jun 25 2012
System image file is "flash:c3900e-universalk9-mz.SPA.151-4.M4.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command
```

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wvl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.

```
Cisco CISCO3945-CHASSIS (revision 1.0) with C3900-SPE250/K9 with 2064384K bytes of
memory.
Processor board ID FTX1541A032
4 Gigabit Ethernet interfaces
2 Voice FXS interfaces
DRAM configuration is 72 bits wide with parity enabled.
256K bytes of non-volatile configuration memory.
500472K bytes of ATA System CompactFlash 0 (Read/Write)
```

License Info:

License UDI:

```
-----
Device#   PID                SN
-----
*0        C3900-SPE250/K9    FOC15391VLH
```




Technology Package License Information for Module:'c3900e'

Technology	Technology-package Current	Technology-package Type	Technology-package Next reboot
ipbase	ipbasek9	Permanent	ipbasek9
security	None	None	None
uc	uck9	Permanent	uck9
data	None	None	None

Configuration register is 0x2102

Configuration:

```
CUBE#sho run
Building configuration...
```

```
Current configuration : 14018 bytes
!
! Last configuration change at 19:28:57 UTC Mon Jun 18 2012 by administrator
! NVRAM config last updated at 19:28:59 UTC Mon Jun 18 2012 by administrator
! NVRAM config last updated at 19:28:59 UTC Mon Jun 18 2012 by administrator
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname CUBE
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.151-4.M4.bin
boot-end-marker
!
!
logging buffered 51200 warnings
enable secret 5 $1$1bTj$25nxZBh5UBjnxOrx9aRvn/
!
no aaa new-model
!
no ipv6 cef
ip source-route
!
!
ip cef
!
!
!
ip domain name yourdomain.com
ip name-server 10.85.0.232
multilink bundle-name authenticated
!
!
!
!
crypto pki token default removal timeout 0
```



```

!
crypto pki trustpoint TP-self-signed-3709846528
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-3709846528
  revocation-check none
  rsakeypair TP-self-signed-3709846528
!
!
crypto pki certificate chain TP-self-signed-3709846528
certificate self-signed 01
  3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 33373039 38343635 3238301E 170D3131 31303039 30303538
  34325A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 37303938
  34363532 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  8100AF32 1ED94037 AE623623 EA4B82FF 75A7D07B 5DDE35FC E43626A7 42B39B06
  25C99F5B C7B03256 9D971028 F863E825 47C7FF04 9DCD132D B826F0D5 883321F8
  9E1E0D66 69800B78 EE313C88 1B1E2650 0FBE01CA B6B4EF52 0FE02FD0 4D8CEFFA
  AEB71024 4E7B426E E5C5BA8F E0D61A9B 06411555 6BFF0236 A7937023 D54D19A8
  D6CB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
  551D2304 18301680 143E4473 FBF89536 7F3209E9 929471C9 60D35F0C FF301D06
  03551D0E 04160414 3E4473FB F895367F 3209E992 9471C960 D35F0CFF 300D0609
  2A864886 F70D0101 05050003 818100A6 083C70BD 5A602FFC 7BBE8F60 49621144
  7E0B6CED 59C85C94 03D11568 7BFA9283 F9C7C5AF AE34CA8E 04263221 61D59D86
  C3AB236D C813EA4B A40CC119 4A953EF9 FB99B4B3 1D8C83E0 60A88B90 51DA70E9
  527B3142 438E17FA 3B15D928 04B00A21 C90DCB98 DDD06C8E 989EBD23 313DC9BE
  651B1988 1B980074 0AF355C2 7381E5
      quit
voice-card 0
  dspfarm
  dsp services dspfarm
!
!
!
voice service voip
  ip address trusted list
  ipv4 10.10.0.133
  ipv4 10.64.1.72
  ipv4 10.70.1.2
  ipv4 200.50.67.0 255.255.255.0
  ipv4 0.0.0.0 0.0.0.0
  ipv4 198.147.226.0
  ipv4 198.147.226.0 255.255.255.0
  address-hiding
  mode border-element
  allow-connections sip to sip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
  rellxx supported "rell100"
  min-se 180 session-expires 180
  g729 annexb-all
!
voice class codec 1
  codec preference 1
  g729br8

```

Enabling CUBE mode

For Fax protocol tests – t38 and g711u protocols were tested. To configure for G711u fax :use command : fax protocol pass-through g711ulaw

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

voice class codec 1
codec preference 1
g729br8

Codecs used for test: Change order of preference based on codec being tested



```
codec preference 2
g711ulaw
codec preference 3
g729r8
```

```
!
!
!
!
voice translation-rule 1
 rule 1 /\+\(1.....\) / /\1/
!
!
voice translation-rule 4
 rule 1 /412142425996/ /12142425996/
!
!
voice translation-profile addplus
 translate called 2
!
voice translation-profile rmv_one
 translate called 3
!
voice translation-profile strip4
 translate called 4
!
!
license udi pid C3900-SPE250/K9 sn FOC15391VLH
!
!
hw-module pvdm 0/0
!
username administrator password 7 105A0C123346085A5C0A
!
redundancy
!
!
!
!
```

```
translation-
rule 2
 Rule 1
6049380601 2629
 Rule 2
6049380602 2700
 Rule 3
6049380603 2644
 Rule 4
6049380604 2604
 Rule 5
6049380605 2302
```

Translation rule for 10 digit
(from MetaSwitch) to 4 digit
private extensions on Cisco.

```
translatio
n-rule 3
 Rule 1
0601 2629
 Rule 2
0602 2700
```

Translation rule for 4 digit
(from MetaSwitch) to 4 digit
private extensions on Cisco.



```
Rule 3
0603 2644
Rule 4
0604 2604
```

```
!
!
```

```
translation-rule
4
Rule 1 3001
6049380601
Rule 2 3002
6049380602
Rule 3 3003
6049380603
Rule 4 3004
6049380604
```

Translation rule for translating
4 digit to 10 digit numbers.

```
!
!
!
!
```

```
interface GigabitEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-
INFO-GE 0/0$
ip address 10.70.10.15 255.255.0.0
duplex full
speed 100
```

CUBE LAN side connection to
CUCM

```
!
```

```
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
```

```
!
```

```
interface GigabitEthernet0/2
description Connection to
Metaswitch
ip address 174.46.0.147
255.255.255.128
duplex full
speed 100 duplex auto
speed auto
```

CUBE WAN side connection to
MetaSwitch

```
!
```

```
interface GigabitEthernet0/3
no ip address
shutdown
duplex auto
speed auto
```

```
!
```

```
ip forward-protocol nd
```

```
!
```

```
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
```

```
!
```



```
ip route 0.0.0.0 0.0.0.0 174.46.0.129
ip route 10.10.0.0 255.255.0.0 192.168.100.1
ip route 10.64.0.0 255.255.0.0 10.70.10.1
ip route 10.70.0.0 255.255.0.0 10.70.10.1
!
access-list 23 permit 10.10.10.0 0.0.0.7
!
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
!
voice-port 0/0/0
  station-id name test
  station-id number
  16049380602
  caller-id enable
!
voice-port 0/0/1
  station-id name test
  station-id number
  16049380604
  caller-id enable
!
!
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 10.70.20.2 identifier 1 priority 1 version 6.0
!
sccp ccm group 23
  description tvgl MTP
  bind interface GigabitEthernet0/0
  associate ccm 1 priority 1
  associate profile 23 register tvglmtp
!
sccp ccm group 111
  description tvgl transcoding
  bind interface GigabitEthernet0/0
  associate ccm 1 priority 1
  associate profile 111 register tvgltxcode
!
dial-peer voice 5996 voip
  shutdown
  destination-pattern 21424259..
  session protocol sipv2
  session target ipv4:10.64.1.72:5060
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  codec g711ulaw
!
```

```
dial-peer voice 9 pots
  description to fax port
  destination-pattern
  6049380602
  no digit-strip
```

Voice Ports configured for
FAX tests – CPE-CPE

Dial peers for FAX call routing



```
port 0/0/0
!  
dial-peer voice 12 pots  
description to fax port  
destination-pattern  
6049380604  
no digit-strip  
port 0/0/1
```

```
!  
dial-peer voice 15 voip  
description Trunk to Metaswitch  
destination-pattern 1.....  
session protocol sipv2  
session target  
dns:whistler.datcon.co.uk  
incoming called-number .%  
voice-class codec 1 offer-all  
dtmf-relay rtp-nte
```

Dial peer for routing all incoming calls towards MetaSwitch

```
!  
dial-peer voice 16 voip  
description To Cluster4  
destination-pattern  
604938060[0-9]  
translate-outgoing called  
2  
session protocol sipv2  
session target  
ipv4:10.70.19.3  
incoming called-number .%  
voice-class codec 1 offer-all  
dtmf-relay rtp-nte
```

Dial peer for routing calls towards CUCM

```
!  
dial-peer voice 17 voip  
destination-pattern 1866.....  
session protocol sipv2  
session target  
dns:whistler.datcon.co.uk  
voice-class codec 1 offer-all  
dtmf-relay rtp-nte
```

Dial peer for routing early media number – United Airlines number

```
!  
dial-peer voice 18 voip  
description to call PBX-PBX  
translation-profile outgoing  
addplus  
destination-pattern 604938060[0-9]  
session protocol sipv2  
session target  
dns:whistler.datcon.co.uk  
voice-class codec 1 offer-all  
dtmf-relay rtp-nte
```

Dial peer for routing calls from CPE-CPE – via MetaSwitch

```
!  
dial-peer voice 19 voip  
description Mapping to private extensions  
shutdown  
destination-pattern 060[0-9]  
translate-outgoing called 3
```



```
session protocol sipv2
session target ipv4:10.70.19.3
incoming called-number 3...
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
dial-peer voice 20 voip
description mapping 4 digit to private
destination-pattern 3...
translate-outgoing called 4
session protocol sipv2
session target dns:whistler.datcon.co.uk
incoming called-number 3...
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
```

```
dial-peer voice 22 voip
description Intl calls
destination-pattern
0119180.....
session protocol sipv2
session target
dns:whistler.datcon.co.uk
incoming called-number .%
voice-class codec 1 offer-all
dtmf-relay rtp-nte
!
```

Dial peer for routing
international calls – that are
identified by a 011 prefix.

```
dial-peer voice 602 voip
description formidcall neg
huntstop
shutdown
destination-pattern 6049380602
translate-outgoing called 2
session protocol sipv2
session target ipv4:10.70.19.3
incoming called-number .%
dtmf-relay rtp-nte
codec g729br8
!
```

```
!
sip-ua
!
```

```
!
gatekeeper
shutdown
!
```

```
!
banner exec ^CC
% Password expiration warning.
```

Cisco Configuration Professional (Cisco CP) is installed on this device and it provides the default username "cisco" for one-time use. If you have already used the username "cisco" to login to the router and your IOS image supports the "one-time" user option, then this username has already expired. You will not be able to login to the router with this username after you exit this session.



It is strongly suggested that you create a new username with a privilege level of 15 using the following command.

```
username <myuser> privilege 15 secret 0 <mypassword>
```

Replace <myuser> and <mypassword> with the username and password you want to use.

```
-----
Feb 23 20:43:16.176: rtpspi_process_timers: Timer 0xA93F798 expired.
Feb 23 20:43:16.176: rtpspi_process_timers: timer expired for type(0x6):
RTP_GLOBAL_INACTIVITY_TIMER
Feb 23 20:43:16.176: rtpspi_handle_rtp_global_timer_expiry: Entered
Feb 23 20:43:16.176: rtpspi_handle_rtp_global_timer_expiry:
rtpspi_count_ccb_with_rtptimeout:0 Nothing to do
```

```
Feb 23 20:43:16.176: rtpspi_start_rtp_global_timer -----
--
^C
banner login ^CC
```

```
-----
Cisco Configuration Professional (Cisco CP) is installed on this device.
This feature requires the one-time use of the username "cisco" with the
password "cisco". These default credentials have a privilege level of 15.
```

YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN CREDENTIALS

Here are the Cisco IOS commands.

```
username <myuser> privilege 15 secret 0 <mypassword>
no username cisco
```

Replace <myuser> and <mypassword> with the username and password you want to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to <http://www.cisco.com/go/ciscocp>

```
-----
^C
!
line con 0
  login local
line aux 0
line vty 0 4
  exec-timeout 0 0
  privilege level 15
  logging synchronous
  login local
  transport input telnet ssh
line vty 5 15
  exec-timeout 0 0
  privilege level 15
  logging synchronous
  login local
  transport input telnet ssh
```




```
!  
scheduler allocate 20000 1000  
end
```



Configuring the Cisco Unified Communications Manager

CUCM Version

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator Search Documentation About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Help

Cisco Unified CM Administration
System version: 8.6.1.20000-1



SIP Trunk Profile:

Figure 2 Non-Secure SIP Trunk Profile

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. The configuration page includes a status section showing "Status: Ready" and a main configuration section titled "SIP Trunk Security Profile Information".

SIP Trunk Security Profile Information

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Stri
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application Level Authorization	
<input type="checkbox"/> Accept Presence Subscription	
<input type="checkbox"/> Accept Out-of-Dialog REFER**	
<input type="checkbox"/> Accept Unsolicited Notification	
<input type="checkbox"/> Accept Replaces Header	
<input type="checkbox"/> Transmit Security Status	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

Buttons: Save, Delete, Copy, Reset, Apply Config, Add New

Footnote: * - indicates required item.
**If this profile is associated with an EMCC SIP trunk, Accept Out-of-Dialog REFER is enabled regardless of the setting on this page



SIP Profile:

Figure 3 SIP profile: Standard SIP Profile- Early Offer (1/2)

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP profile. The page title is "SIP Profile Configuration" and the profile name is "Standard SIP Profile -early offer". The status is "Ready". The configuration is divided into three main sections: SIP Profile Information, Parameters used in Phone, and a list of checkboxes for various features.

Section	Field/Option	Value
SIP Profile Information	Name *	Standard SIP Profile -early offer
	Description	Default SIP Profile
	Default MTP Telephony Event Payload Type *	101
	Resource Priority Namespace List	< None >
	Early Offer for G.Clear Calls *	Disabled
	SDP Session-level Bandwidth Modifier for Early Offer and Re-invites *	TIAS and AS
	User-Agent and Server header information *	Send Unified CM Version Information as User-Ager
	<input type="checkbox"/> Redirect by Application	
	<input type="checkbox"/> Disable Early Media on 180	
	<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
Parameters used in Phone	Timer Invite Expires (seconds) *	180
	Timer Register Delta (seconds) *	5
	Timer Register Expires (seconds) *	3600
	Timer T1 (msec) *	500
	Timer T2 (msec) *	4000
	Retry INVITE *	6
	Retry Non-INVITE *	10
	Start Media Port *	16384
	Stop Media Port *	32766
	Checkboxes	<input type="checkbox"/> Enable ANAT
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change		
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests		
<input type="checkbox"/> Allow Presentation Sharing using BFCP		



Figure 4 Non-Secure SIP Trunk Profile (2/2)

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SIP Profile Configuration

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Save Delete Copy Reset Apply Config Add New

Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

Conference Join Enabled
 RFC 2543 Hold
 Semi Attended Transfer
 Enable VAD
 Stutter Message Waiting

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

RSVP Over SIP* Local RSVP

Fall back to local RSVP

SIP Rel1XX Options* Disabled

Deliver Conference Bridge Identifier
 Early Offer support for voice and video calls (insert MTP if needed)
 Send send-receive SDP in mid-call INVITE

SIP OPTIONS Ping

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6



Trunk to CUBE:

Figure 5 Trunk to CUBE (1/3)

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the status is "Ready". The configuration is for a device named "CUBE".

Field	Value
Product	SIP Trunk
Device Protocol	SIP
Trunk Service Type	None(Default)
Device Name*	CUBE
Description	CUBE
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default



Figure 6 Trunk to CUBE (2/3)

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Save Delete Reset Add New

Route Class Signaling Enabled* Default

Use Trusted Relay Point* Default

PSTN Access

Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* 4

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>



Figure 7 Trunk to CUBE (3/3)

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Trunk Configuration Related Links: [Back To Find/List](#)

Save Delete Reset Add New

Connected Party Settings

Connected Party Transformation CSS

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port
1* <input type="text" value="10.70.10.15"/>	<input type="text"/>	<input type="text" value="5060"/>

MTP Preferred Originating Codec*

Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

Parameter Name	Parameter Value
----------------	-----------------

Configuring Route Patterns

Route patterns are configured as below, prefixed by a 4 to identify that a call has to be routed via the CUBE.

<input type="checkbox"/>	4.0119180XXXXXXXX	For International Calls via MetaSwitch	CUBE	<input type="button" value="🔗"/>
<input type="checkbox"/>	4.1XXXXXXXXXX	TO CUBE for MetaSwitch for PSTN calls	CUBE	<input type="button" value="🔗"/>
<input type="checkbox"/>	4.300X	TO CUBE for MetaSwitch	CUBE	<input type="button" value="🔗"/>
<input type="checkbox"/>	4.604938060X	TO CUBE for MetaSwitch	CUBE	<input type="button" value="🔗"/>



Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
SCCP	Skinny Client Control Protocol
Cisco UCM	Cisco Unified Communications Manager
Cisco UBE	Cisco Unified Border Element
PSTN	Public Switched Telephone Network



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Printed in the USA